packets at the same rate. To prevent the performance penalty caused by this situation, you can now configure the system not to send ICMP Destination Unreachable packets.

Caution: This feature is intended only for VoIP environments. Enabling this feature can break required behavior for IPv4 routers, such as Path MTU Discovery.

Following is the relevant parameter, shown with its default setting:

```
[in IP-GLOBAL]
send-icmp-dest-unreachable = yes
```

Parameter	Specifies
Send-ICMP-Dest-Unreachable	Enable/disable sending of ICMP Destination Unreachable packets. The default is `yes`. If set to `no`, the MAX TNT does not send ICMP Destination Unreachable packets. Setting this parameter to No is recommended only for VoIP environments.

The following commands disable transmission of ICMP Destination Unreachable packets:

```
admin> read ip-global
IP-GLOBAL read

admin> set send-icmp-dest-unreachable = no

admin> write
IP-GLOBAL written
```

Preventing receipt of UDP packets until VoIP calls are set up

When two MultiVoice Gateway systems are establishing the link for transmission of a VoIP call, both systems do not always complete the call setup at the same time. However, a Gateway starts sending UDP packets to the other Gateway as soon its own call setup is complete. If the receiving Gateway has not yet set up its port caches, the shelf controller receives the UDP packets for a period of time until the call is fully set up. Now, you can prevent receipt of UDP packets until the link is fully established. Following is the relevant parameter, shown with the default value:

```
[in IP-GLOBAL]
throttle-no-port-match-udp-traffic-on-slot = no
```

Parameter	Specifies
Throttle-No-Port-Match-UDP-Traffic-On-Slot	Enable/disable reception of UDP packets for UDP ports currently unknown to the MAX TNT. With the default value of `no`, the system behaves as in previous releases and sends the unknown port packets to the shelf controller for processing. If the parameter is set to `yes`, the system discards UDP packets until the UDP port is known. The setting of `yes` is recommended for MultiVoice Gateways, to prevent overloading of the shelf controller when both Gateways do not always complete the VoIP call setup at the same time.

The following commands enable the system to discard UDP packets until the UDP port is known:

```
admin> read ip-global
IP-GLOBAL read
```

```
admin> set throttle-no-port-match-udp-traffic-on-slot = yes

admin> write
IP-GLOBAL written
```

System settings for VoIP operations

Lucent recommends setting the following parameters, shown with default values, to facilitate VoIP call handling:

```
[in IP-GLOBAL]
system-ip-addr = 0.0.0.0

[in ANSWER-DEFAULTS:session-info]
idle-timer = 0

[in SYSTEM]
max-dialout-time = 60
parallel-dialing = 32
country = us
```

Parameter	Recommended VoIP settings
System-IP-Addr	In an H.323 environment, set this parameter to the shelf controller IP address. In an IPDC environment, if the system allocates its own listen address, set this parameter to the IP address of a LAN interface other than the shelf controller port.
Idle-Timer	For real-time fax or transparent modem calls, set this parameter should be set to 0 to disable the idle timer and prevent the fax or modem calls from timing out.
Max-Dialout-Time	To allow sufficient time for the MAX TNT to establish the connection to the called destination, and for consistency with internal H.323 timers, a setting of 60 is recommended.
Parallel-Dialing	To decrease the instances when VoIP callers wait for a silent interval while the MAX TNT completes a call that has been queued, a setting of 32 is recommended.
Country	Setting this parameter to the appropriate value enables the MAX TNT to generate country-specific local call-progress tones (such as dial tone, busy signals, and so forth), based on the ITU-T specification TSB Circular 18: *Update of Supplement No. 2, ITU-T (former CCITT) Blue Book, Fascicle II.2 - Various tones used in national networks.* The following country-specific call progress tones are currently supported by MultiVoice: Argentina, Australia, Belgium, China, Costa Rica, Finland, France, Germany, Hong Kong, Italy, Japan, Korea, Mexico, Netherlands, New Zealand, Singapore, Spain, Sweden, Switzerland, United Kingdom, and the United States (the default).

For example, the following commands configure the system in a recommended way for VoIP call handling:

```
admin> read answer-defaults
ANSWER-DEFAULTS read

admin> set session-info idle-timer = 0

admin> write
ANSWER-DEFAULTS written
```

```
admin> read system
SYSTEM read
admin> set max-dialout-time = 60
admin> set parallel-dialing = 32
admin> set country = us
admin> write
SYSTEM written
```

Call route configuration

The MAX TNT Release 8.0-103 supports simultaneous processing of voice and data calls. To simplify routing of voice and data traffic between the MAX TNT and PSTN:

- Use DNIS-specific trunk mappings for detection of voice calls
- Process data and voice calls on different MultiDSP cards
- Use preferred source routing method (optional) for data call types
- Use trunk routing (optional) for outbound voice calls

Using DNIS-specific trunk mappings

The default VoIP profile, `voip { 0 0 }`, is a system-wide profile used for processing all VoIP calls. Additional VoIP profiles may be created to simplify processing and routing of VoIP calls.

User-defined VoIP profiles are used to map incoming calls by identifying all calls associated with a specific Dialed Number Identification Service (DNIS) string as VoIP calls. See "Creating user defined VoIP profiles" in the *MultiVoice for the MAX TNT Configuration Guide* for details.

Examples of user defined VoIP profiles

For example, if a user created the following VoIP profiles:

```
admin> dir voip
   46  12/23/1998 09:48:55  { 0 0 }
   31  12/18/1998 09:50:06  { 8093190 0 }
   31  12/18/1998 10:07:16  { 8903190 0 }
```

The MAX TNT will process all calls from the PSTN with these DNIS strings as VoIP calls. The Voip-Index subprofile distinguishes between the default VoIP profile, `voip { 0 0 }`, and any user created VoIP profiles:

```
admin> list voip-index
[in VOIP/{ 8903190 0 }:voip-index
gateway-access-number = 8903190
far-end-number = 0
```


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This subprofile includes the following parameters:.

Parameter	Specifies
Gateway-Access-Number	This is the Dialed Number Identification String (DNIS) passed from the PSTN associated with the in-bound telephone number used to access the MAX TNT. If the MAX TNT is configured to perform two-stage dialing of VoIP calls, this would be the telephone number dialed to access the MAX TNT from the PSTN.
Far-End-Number	This value should always be set to `0`.

Note: This modification is made after the MAX TNT is initialized. Once these changes are committed, save the new configuration to flash memory or a tftp server. The saved image may be retrieved to restore this configuration in the event that a MAX TNT must be re-initialized.

Process voice and data calls on different MultiDSP cards

To enable the simultaneous processing of voice and data calls, you must create exclusive call routing types for each MultiDSP card. This is accomplished by deleting the Call-Route profiles for call types which should not be accepted for processing by a MultiDSP card.

At startup, up to four default Call-Route profiles are automatically created to handle different call types. Hash codes on the shelf controller determine which call route type profiles are created. The MAX TNT uses this profile to control which calls are accepted and processed by each MultiDSP card. Every possible destination within a MAX TNT system has one or more profiles of this type.

At least four types of Call-Route profiles are created for each installed MultiDSP card, as illustrated in the following `callroute` command output:

```
admin>callroute -d
device      #  source       type               tg   sa  phone
1:03:01/0   1  0:00:00/0   digital-call-type   0    0
1:03:01/0   2  0:00:00/0   phs-call-type       0    0
1:03:01/0   3  0:00:00/0   voip-call-type      0    0
1:03:01/0   4  0:00:00/0   v110-call-type      0    0
```

The supported profile types for the MultiDSP card include:

Type	Description
Digital-Call-Type	General digital calls, including 3.1 Khz audio bearer channel calls can be routed to a device with this call route type. This is a host device. This Call-Route profile has an index of 1.
Phs-Call-Type	PHS calls can be routed to a device with this call route type. This Call-Route profile has an index of 2.
Voip-Call-Type	VOIP calls can be routed to a device with this call route type. This Call-Route profile has an index of 3.

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Type	Description
V110-Call-Type	Digital calls recognized as containing V.110 rate adapted bearer channels cat be routed to device with this call route type. This Call-Route profile has an index of 4.

Depending upon whether the MultiDSP card will process voice or data calls, you should delete the call types as listed in the following:

For this default call type	Delete the following Call-Route profiles
VoIP calls (voip-call-type)	Any-Call-Type, Digital-Call-Type, V110-Call-Type
Data calls (digital-call-type)	Any-Call-Type, Voip-Call-Type

For example, if a MultiDSP card should only process VoIP calls, you would delete the Digital-Call-Type, V110-Call-Type and the Any-Call-Type profiles for the selected MultiDSP card.

Note: For all locations except Japan, the Phs-Call-Type Call-Route profile need not be deleted for MultiDSP cards processing voice calls. Currently, PHS calls are only supported by PSTNs in Japan.

To remove Call-Route profiles execute the following:

1 Use the show command to identify all the MultiDSP (madd-card) cards installed in your MAX TNT:

```
admin>show
Shelf 1 ( standalone ):
{ shelf-1 slot-1 0 }          UP       8e1-card
{ shelf-1 slot-2 0 }          UP       ether3-card
{ shelf-1 slot-3 0 }          UP       madd-card
{ shelf-1 slot-4 0 }          UP       madd-card
{ shelf-1 slot-5 0 }          UP       madd-card
{ shelf-1 slot-6 0 }          UP       madd-card
{ shelf-1 slot-7 0 }          UP       madd-card
{ shelf-1 slot-8 0 }          UP       madd-card
admin>
```

2 Delete the Call-Route profiles for each call type a MultiDSP card should not accept. To delete the Call-Route profile for V110-Call-Type processing on the MultiDSP slot card in slot 3, execute the following command:

```
admin> delete call-route { { {1 3 0} 0} 4}
Delete profile CALL-ROUTE/{ { { shelf-1 slot-3 0 } 0 } 4 }? [y/n] y
CALL-ROUTE/{ { { shelf-1 slot-3 0 } 0 } 4 } deleted
admin>
```

Repeat this procedure for each Call-Route profile associated with an excluded call type.

Note: This modification is made after the MAX TNT is initialized. Once these changes are committed, save the new configuration to flash memory or a tftp server. The saved image may be retrieved to restore this configuration in the event that a MAX TNT must be re-initialized.

Configuring preferred source routing

Using preferred source routing configures the MAX TNT to direct calls from the designated network device, (such as, T1 or E1 slot cards) to a specific MultiDSP card. This may be used to limit the calls a MultiDSP card accepts for processing to a specific T1 or E1 channel, and may be used for routing data calls.

This is accomplished by assigning the address of a T1 or E1 channel to the Preferred-Source parameter in the Call-Route profiles for each data call type configured for a MultiDSP card. This address identifies the shelf, slot, and connection associated with a specific T1 or E1 trunk.

To configure preferred source routing, execute the following:

1 Use the `show` command to identify all the T1 or E1 cards installed in your MAX TNT:

```
admin>show
Shelf 1 ( standalone ):
{ shelf-1 slot-1 0 }        UP        8e1-card
{ shelf-1 slot-2 0 }        UP        ether3-card
{ shelf-1 slot-3 0 }        UP        madd-card
{ shelf-1 slot-4 0 }        UP        madd-card
{ shelf-1 slot-5 0 }        UP        madd-card
{ shelf-1 slot-6 0 }        UP        madd-card
{ shelf-1 slot-7 0 }        UP        madd-card
{ shelf-1 slot-8 0 }        UP        madd-card
admin>
```

2 For each MultiDSP card, change the value assigned to the Preferred-Source parameter in the Call-Route profile for Digital-Call-Type. To route calls received through any E1 connected on slot 1 to the MultiDSP card in slot 4, execute the following command:

```
admin> read call-route { { {1 4 0} 0} 1}
CALL-ROUTE/{ { { shelf-1 slot-4 0 } 0 } 1 } read

admin>set preferred-source={{1 1 0} 0}

admin>write
CALL-ROUTE/{ { { shelf-1 slot-4 0 } 0 } 1 } written

admin>
```

You may configure a routing using all the T1 or E1 connections on the ingress card, as in the example, or specify an individual trunk by identifying a specific port on the ingress card, for example:

```
admin>set preferred-source={{1 1 4} 0}
```

Repeat this procedure until all T1 or E1 trunks are mapped to MultiDSP cards.

Note: This modification is made after the MAX TNT is initialized. Once these changes are committed, save the new configuration to flash memory or a tftp server. The saved image may be retrieved to restore this configuration in the event that a MAX TNT must be re-initialized.

Use trunk routing (optional) for outbound voice calls

Trunk routing of outbound VoIP calls is used to control allocation of T1 or E1 trunks for voice calls. The MAX TNT, which connects a VoIP call to the destination telephone number, can automatically route calls to the PSTN using a trunk group selected by the MAX TNT which initiated the call.

To utilize automated trunk routing:

- Trunk groups must be enabled on both MAX TNTs used to connect the call

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- Both MAX TNTs should have the same number of T1 or E1 trunks available for connecting VoIP calls
- Both MAX TNTs must utilize the same trunk numbering scheme

When trunk prefixing is enabled, the MAX TNT obtains the trunk group number of the ingress T1 trunk from the `trunk-group` setting in the T1 line profile, and prefixes it to the detected DNIS, the destination telephone number. The MAX TNT modifies Q.931 Called Party Number information element (IE) in an H.225/Q.931 SETUP message, sending the DNIS number prefixed by the incoming trunk number to the MAX TNT which connects the voice call.

When the destination MAX TNT dials the call, it will connect the call to the PSTN using a trunk assigned to the requested trunk group.\

Enabling trunk groups

To enable automated trunk group processing of VoIP calls, you must configure the following:

Parameter	Profile	Value(s)	Description
Use-Trunk-Groups	System	Yes	This parameter enables the use of trunk groups for all network lines. When this parameter is enabled, all channels must be assigned a trunk group number for outgoing calls.
Num-Digits-Trunk-Groups	System	1- 4	This parameter sets a limit of the number of digits that may be used to designate trunk groups. The value assigned this parameter limits size of the values assigned to the Trunk-Group parameter to one- through four-place numbers.
Trunk-Group	T1 or E1	1 - 9999	This parameter assigns a channel to a trunk group. In a T1 or E1 profile, the default is 9. Individual channels may be assigned to different trunk groups.
Trunk-Prefix-Enable	Voip { x x }	Yes	This parameter enables outbound routing of VoIP calls over trunk groups from the ingress MAX TNT. The ingress MAX TNT will send the trunk group address as part of the dial string for the destination telephone number.

VoIP call management and performance settings

In an H.323 environment, settings in the default VoIP profile are used for processing all VoIP calls. In an SS7 environment, settings in the default VoIP profile are used only for settings that are not superseded by values in IPDC messages.

For details about VoIP profile settings that are new in MAX TNT TAOS 8.0-103, see *New VoIP profile settings in MAX TNT TAOS 8.0-103* on page 23.

New VoIP profile settings in MAX TNT TAOS 8.0-103

The following parameters (shown with default values) are new or modified in MAX TNT TAOS 8.0-103:

```
[in VOIP/{ 0 0 }]
gk-mlg-control = no
signaling-model = early-alerting

[in VOIP/{ 0 0 }:rt-fax-options]
packet-redundancy = no
fixed-packets = yes
max-rate = 9600
```

Parameter	Specifies
Gk-Mlg-Control	The Gk-Mlg-Control parameter enables the MultiVoice Gateway to accept and process call-specific administration instructions from a MultiVoice Access Manager, Release 3.0. When enabled, the gateway may apply call-specific processing instructions, for PIN authentication, single- or two-stage dialing, voice announcement playback, and configuring call timers for pre-paid billing. Values received from MVAM, or a third party billing system, will override parameter settings in the Voip { X X } profile for processing the current VoIP call. Rules used for performing call-specific administration are configured on MVAM, and are used when partitioning MultiVoice Gateways into multiple logical gateways. This allows MVAM to administer a single physical gateway as if it were multiple gateways, partitioning the gateway according to trunk groups, DNIS, time of day, etc.
Signaling-Model	This parameter controls processing of the H.245 startup procedure, by defining the relationship between H.323 alert messaging and PSTN alerting. This parameter creates a virtual inband pipeline for call signal processing by mapping PSTN actions into H.323 actions. When enabled, the H.245 startup information delivered in the Call Proceeding message provides for more transparent PSTN signaling behavior.
Packet-Redundancy	This parameter sets the packet redundancy scheme and jitter buffering for Multivoice Real-time fax over un-managed networks (such as, the public internet). When enabled, this parameter causes a MAX TNT to append the designated number of previously sent fax packets onto the current packet. On networks experiencing measurable packet loss, this improves the reliability of the fax transmission.
Fixed-Packets	This parameter lets customers disable the jitter buffer and packet redundancy scheme for Real-time fax calls. When packet redundancy is disabled, a MultiVoice Gateway running a pre-8.0-103 software release can process Real-time fax calls to and from a MultiVoice Gateway running the 8.0-103 software.

Parameter	Specifies
Max-Rate	This parameter sets the maximum data transmission rate allowed for a T.38 fax session configurable on a MultiVoice Gateway. This provides customers with a means to regulate the bandwidth used for fax sessions on their networks.

Configuring multiple logical gateways (MLG)

Using gatekeeper controlled multiple logical gateways is a method of performing call-specific administration of H.323 VoIP calls. The following call control functions may be used for partitioning one physical MultiVoice Gateway into multiple logical gateways from the gatekeeper:

- PIN prompting
- Single-stage dialing
- Two-stage dialing
- Voice announcement playback
- Configurable call timers for pre-paid and credit card billing systems

The MultiVoice Access Manager (MVAM) analyzes call performance data (trunk group, ds0 status and call activity), received when a gateway performs periodic keep-alive registration. When MVAM responds to subsequent call requests from each gateway, the Admission Conformation (ACF) message will include any changes defined for the aforementioned call administration parameters. The gateway applies the parameter changes received from MVAM to the current call request. This information is stored as part of the non-standard data included in registration, admission and status (RAS) messages exchanged by the gateway and gatekeeper for each call.

Warning: Dynamic call control and multiple logical gateways are only supported in Multi-Voice networks running TAOS Release 8.0-103 on the gateways and MVAM Release 3.0 on the gatekeepers. These features are not supported in MultiVoice networks where gatekeepers are running Release 2.x of the MultiVoice Access Manager.

Previously, all H.323 call management features were configured globally, on each MultiVoice Gateway, using the values assigned in the VOIP Options profile. Now, utilizing status information reported by MultiVoice Gateways, a gatekeeper running MultiVoice Access Manager, Release 3.0 may send instructions to the ingress gateway which override global call management settings. The decision to override the global call management settings may be based upon reported ingress trunk or DS0 groups, Caller ID, time-of-day, gateway, etc.

The rules used to apply overrides to H.323 call management parameters are configured on MVAM. These parameter changes are useful when partitioning MultiVoice Gateways into logical gateways. *Logical gateways*, defined on MVAM, treat selected trunk groups on a MultiVoice Gateway as if they were a unique VoIP gateway. Initially, MultiVoice Gateways must have T1, T3 and PRI trunks to support logical gateways. A MultiVoice Gateway won't know about its logical gateways, only MVAM does. However, a gateway must be configured to apply instructions received from MVAM when processing the current call.

Note: While BRI lines may still be used for VoIP, the multiple logical gateway features are not supported on MultiVoice Gateways using BRI.

MVAM may enable call-specific administration based upon the reported DNIS, ANI, trunk group and DS0 information, or any combination of that data, which are all reported in the first ARQ from the gateway.

Dynamic PIN authentication

When the multiple logical gateway feature is enabled on a MultiVoice Gateway, any incoming call request will immediately send an ARQ to MVAM which includes:

- DNIS, when available
- ANI, when available
- Trunk group and DS0 status changes

If the ARQ includes all the information necessary to route the call, MVAM will send an ACF message to the gateway. The gateway will then process the call as if the following VoIP parameters were set to these values:

```
vpn-mode=yes
single-dial-enable=yes
```

If MVAM, or a third party billing application used with MultiVoice, requires PIN authentication for this call, an Admission Reject (ARJ) message is issued directing the gateway to set `vpn-mode=no` for this call. The gateway will then resume call handling as if the call had just arrived from the PSTN, but prompt for authentication (as if `vpn-mode=no`) before continuing with call processing.

Dynamic single-stage and two-stage dialing

When the multiple logical gateway feature is enabled on a MultiVoice Gateway, any incoming call request will immediately send an ARQ to MVAM which includes:

- DNIS, when available
- ANI, when available
- Trunk group and DS0 status changes

If the ARQ includes all the information necessary to route the call, MVAM will send an ACF message to the gateway. The gateway will then process the call as if the following VoIP parameters were set to these values:

```
vpn-mode=yes.
single-dial-enable=yes
```

If MVAM, or a third party billing application used with MultiVoice, requires a caller perform two-stage dialing for this call (dialing the destination telephone number after dialing into the MultiVoice Gateway), an Admission Reject (ARJ) message is issued directing the gateway to set `single-dial-enable=no` for the call. The gateway will then resume call handling as if the call had just arrived from the PSTN, but prompt the caller to enter the destination telephone number (`single-dial-enable=no`) before continuing with call processing.

Static announcement branding

When the multiple logical gateway feature is enabled on a MultiVoice Gateway, MVAM, or a third party billing application, may select a set of voice announcements stored for playback from multiple sets of voice announcements stored on the gateway. This is known as *branding*.

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By sending either an ARJ or ACF message, containing an announcement directory specifier, the gateway will playback voice announcements from the named directory on the pc-flash card for the current call.

Executing the branding instructions, the gateway will search for the voice announcement directory using the value assigned to the Voice-Ann-Dir parameter. When `voice-ann-dir=/current` (default), when MVAM requests a specific directory (brand) of announcements for a call, the gateway will search for those announcements starting in the /current directory. For example, if MVAM specified "italian", the gateway would search for announcements in the directory `/current/italian/`.

Note: It is recommended that only four "brands" of static announcements are used, due to limitations in the announcement cache size. Using more than four will degrade announcement quality and overall gateway performance.

Configurable call timers

This release of MultiVoice supports the use of configurable call timers, controlled by MVAM or a third party billing application, which support timed billing plans (such as: pre-paid phone cards, pre-paid cellular accounts).

Using an ACF message, MVAM or a third party billing application, set the following timers:

Timer	Description
Call countdown timer	This timer sets the time remaining before a gateway disconnects the current call. When this timer expires, the gateway will play an announcement that time has expired and disconnects the call. • By default, once the timer is set on the gateway, the `h323drq.au` announcement file is played back for the caller upon call termination • If the MVAM or third party billing application uses its own countdown timer, the announcement specifier in an Disengage Request (DRQ) massage may be used to select a different announcement file for playback upon call termination
Call disconnect warning timer	This timer specifies when a call disconnect warning announcement will be played for the caller. This announcement alerts the caller to the time remaining before this call is terminated. • By default, once this timer is set on the gateway, the `h323bkin.au` announcement file is played back for the caller when this timer expires • If the MVAM or third party billing application uses its own disconnect warning timer, the announcement specifier in an Interrupt Request (IRQ) massage may be used to select a different announcement file for playback when this timer expires

New Trunk and Call status reporting

Each MultiVoice Gateway reports its current call processing status as part of a Registration Request (RRQ) message to MVAM. This message includes data on trunk, trunk group and DS0 status. The initial RRQ message, sent to MVAM when a gateway is initialized, will contain a full report on all the trunks used by the physical gateway. The RRQ messages sent during keep-alive registration include only the status changes since the previous registration message.

H.323 call-specific administration messages

Call administration information is transmitted as part of the non-standard data included in registration, admission and status (RAS) messages exchanged between the gateway and gatekeeper for each call. This data consists of a set of parameters using URL encoding, as described in RFC 1738, with each parameter composed of a set of attribute value pairs.

This non standard data may include the following call administration information:

- ANI/CLID
- Conference identifier
- User PIN
- Inbound or outbound trunk identification
- Enable voice announcement playback
- Select voice announcement playback
- Internal call timer and disconnect timer settings
- Call failures
- Call results
- Trunk group and DS0 status information
- Available digital signal processors (DSPs)
- Maximum number of calls a MultiVoice Gateway may support

DS0 Status (in-service/out-of-service)

A MultiVoice Gateway reports trunk, trunk group, and DS0 information to MVAM for each trunk. This includes:

- Trunk group
- Physical address
- DS0 service status (in-service or out-of-service)

Note: A DS0 is in-service for a logical gateway when it belongs to the associated trunk group and is in the "up" state. Information regarding DS0 activity (in-use, free) is not reported via RRQ. This is handled separately. traced from the per-call trunk/DS0 reporting mentioned below.

Trunk groups and physical address (shelf, slot, etc.) information are provided to MVAM to allow dynamic tracking of DS0 activity and trunk group assignments, and provided for future support of DS0 selection by physical-address for outbound PSTN calls.

Full trunk and DS0 status reporting is performed only when necessary, enhancing gateway performance. Full RRQ's are used to report complete trunk and DS0 information, usually when a gateway is initialized or else when requested by MVAM. Lightweight RRQ's are used to

report only status changes for trunk and DS0 information. MVAM may request complete trunk and DS0 information by responding to a lightweight RRQ with a Registration Reject (RRJ) message containing a reject reason of `FullRegsitrationRequired`.

Note: Currently, trunk and DS0 status is not reported for BRI lines. Only the following information is reported for MultiVoice Gateways using BRI:

- Number of idle VOIP ports.
- value of maxCalls in VOIP profile.

Trunk and DS0 reporting (per call)

For each call processed by a MultiVoice Gateway, trunk group and physical address information for the DS0 connection are reported. This information is sent from the gateway to the gatekeeper as non-standard data in these registration, admission and status (RAS) messages, for the following call types:

Message	Call type	Trunk or DS0 information
Admission Request (ARQ)	Inbound (from PSTN)	The trunk group and physical address of the DS0 upon which the call arrived.
Bandwidth Request (BRQ)	Outbound (to PSTN)	The trunk group and physical address of the DS0 upon which the call went out.
Disengage Request (DRQ)	Inbound (from PSTN) and Outbound (to PSTN)	The physical address of the DS0. For outgoing PSTN calls, the trunk group or DS0 information may not be present.
Disengage Confirmation (DCF)	Inbound (from PSTN) and Outbound (to PSTN)	The trunk group and DS0 information for gatekeeper-initiated call terminations.

Trunk and DS0 selection (per call)

Currently, MultiVoice Gateways only support trunk-group based routing for outbound PSTN calls. To do this, using trunk groups must be enabled in the System profile of each gateway in the MultiVoice network. Each T1 must also be assigned a trunk group.

Note: Trunk groups should only be assigned at the T1 level.

The physical address information collected by the gateway for each DS0 is used currently by MVAM to dynamically track DS0 activity. It is currently not used for DS0 to DS0 linking. In the future, both trunk group and/or physical address information will be available for DS0 selection on the gateway. When this happens, trunk groups should only be used when processing both VoIP and data calls on the same gateway. Otherwise, only gatekeeper, physical-address based, DS0 routing should be used.

Usage: This feature is enabled or disabled by assigning either `Yes`, enabling processing of call-specific administration instructions, or `No` (default), reverting global administration of VoIP calls using the values set in the Voip { X X } profile.

The following example illustrates how to enable multiple logical gateway processing on this MAX TNT:

```
admin> read voip { 0 0 }
VOIP/{ 0 0 } read

admin> set gk-mlg-control=yes

admin> write
VOIP/{ 0 0 } written
```

Dependencies: This parameter has the following dependencies:

- If `gk-mlg-control=yes`, the value of Vpn-Mode defaults to `N/A`
- If `gk-mlg-control=yes`, the value of Single-Dial-Enable defaults to `N/A`
- Changes to this parameter are effective with the next VoIP call

Location: Voip { X X }

Configuring the H.245 pipeline signal model

By tying the H.323 Alert messaging to PSTN Alerting, the gateway conveys H.245 startup information on top of the Call Proceeding message. This creates a virtual inband pipeline for call signal processing by mapping PSTN actions into H.323 actions.

In all cases, the H.245 connection information is included in all H.323 messages (Call Proceeding, Alerting, and Connect). This enables a gateway to provide the support for the following inband messaging modes:

Inband messaging mode	Description
Early alerting	In this mode, the H.323 Call Proceeding message is sent upon receipt of the Admission Confirmation (ACF) from the gatekeeper, the H.323 Alerting message is sent upon receipt of WAN inband notification from the outdialed trunk, and an H.323 Connect message is sent up receipt of the PSTN Connect message.
Slow proceeding	In this mode, the H.323 Call Proceeding message is sent upon receipt of WAN inband notification from the outdialed trunk, an H.323 Alerting message is sent upon receipt of the PSTN Alerting message, and the H.323 Connect message is sent upon receipt of the PSTN Connect message.
Fast proceeding	In this mode, which is recommended for use over high latency links, the H.323 Call Proceeding message is sent upon receipt of the Admission Confirmation (ACF) message from the gatekeeper, an H.323 Alerting message is sent upon receipt of the PSTN Alerting message, and the H.323 Connect message is sent upon receipt of the PSTN Connect message.

Usage: The Signaling-Model parameter sets the inband messaging mode used by the gateway when mapping H.323 alert messaging and PSTN alerting, and accepts the following values.

Parameter value	Description
`early-alerting`	This value (default) enables inband call signal processing on a gateway using the Early Alerting inband messaging mode.
`slow-proceeding`	This value enables inband call signal processing on a gateway using the Slow Proceeding inband messaging mode.
`fast-proceeding`	This value enables inband call signal processing on a gateway using the Fast Proceeding inband messaging mode.

The following example illustrates how to change the default value of the Signaling-Model parameter.

```
admin> read voip { 0 0 }
VOIP/{ 0 0 } read

admin> set signaling-model=fast-proceeding

admin> write
VOIP/{ 0 0 } written
```

Dependencies: Changes made to the Signaling-Model parameter take effect with the next VoIP call.

Location: Voip { x x }

Enabling fax packet redundancy

Redundant packet data is defined as the last *n* packets transmitted appended to the current packet. The value of *n* is set through the CLI using the Packet-Redundancy parameter. Once defined, this parameter controls processing of several hundred milliseconds of packet jitter and allows the optional transmission of redundant packet data for fax calls across networks experiencing instances of packet loss and packet jitter.

Assigning the Packet-Redundancy parameter a value (such as, `packet-redundancy` = 4), will cause MAX TNT to append that number of previously sent packets onto the current packet. On networks experiencing measurable packet loss, this improves the reliability of the fax transmission.

Depending upon the amount of measurable packet loss for a network, the redundancy parameter should be set accordingly:

Network condition	Recommended value(s)
Packet loss occurs in frequent bursts.	1 - 5
Occasional packet loss (less than one percent)	0 (default)
Occasional packet loss (greater than one percent)	1 - 2

The additional bandwidth required for each fax call increases proportionally to the level of redundancy, adding 50 bytes of packet data per increment. To support this feature, MultiVoice requires Real-time fax support be enabled on the MultiVoice Gateway. This may be verified by checking the Base profile for the `rt-fax-enabled=yes` entry.

This enhancement uses a slip buffer to :

- Allow MultiVoice Real-time fax to tolerate packet jitter
- Keep the modem fed with data, preventing modem underrun

Fixed sized packet format

The packet redundancy scheme uses a fixed-size packet format, consisting of a 49-byte payload, a prefixed sequence number, and a length field which precedes the payload data. When packet redundancy is enabled, *n*-length payload pairs are added at the end of the packet; where *n* is the value of the Packet-Redundancy parameter. Previously, MAX TNT sent variable length packets that were guaranteed to be zero terminated; allowing Class 1 modems to underrun gracefully.

Usage: The Packet-Redundancy parameter accepts values from 0 through 5, directing MultiVoice to append the designated number of previously transmitted fax packets to the current packet, as follows:

Parameter value	Specifies
0	No change from the default packet behavior.
1	Append and send the previous fax packet with the current fax packet.
2	Append and send the two previous fax packets with the current fax packet.
3	Append and send the three previous fax packets with the current fax packet.
4	Append and send the four previous fax packets with the current fax packet.
5	Append and send the five previous fax packets with the current fax packet.

The following example illustrates how to change the default value of the Packet-Redundancy parameter.

```
admin> read voip { 0 0 }
VOIP/{ 0 0 } read

admin> set packet-redundancy=4

admin> write
VOIP/{ 0 0 } written
```

Dependencies: The following dependencies apply to this parameter:

- Once saved, packet redundancy is enabled with the next VoIP call
- This value is set to `N/A` when `fixed-packets=no`.

Location: Voip{x x}>Rt-Fax-Options

Enabling fixed-sized fax packets for backwards compatibility

The Fixed-Packets parameter disables use of redundant packets and the slip buffer for MultiVoice Real-time fax, enabling the pre-8.0-103 release fax packet scheme. When enabled, fax calls are processed using variable length packets that are zero terminated; allowing Class 1 modems to underrun gracefully.

The packet sequence numbering introduced in Release 8.0-103 for Real-time fax required a format change, creating high speed data packets. When these packets are absent (such as, a fax call is initiated from a MultiVoice Gateway running a pre-8.0-103 software release) the MultiVoice Gateway interprets image data as sequence data. Also the smaller packets forwarded by the new code rely on the slip buffer to keep the modem fed with data or it will drop carrier.

Usage: When the value of this parameter is `yes`, the default, the pre-8.0-103 fax packet scheme is enabled. When the value of this parameter is `no`, jitter buffering and packet redundancy for Real-time fax processing is enabled.

The following example illustrates how to enable multiple logical gateway processing on this MAX TNT:

```
admin> read voip { 0 0}
VOIP/{ 0 0} read
admin> set fixed-packets=no
admin> write
VOIP/{ 0 0 } written
```

Dependencies: The following dependencies apply to this parameter:

- Once saved, the selected packeting scheme is enabled with the next fax call
- When this value is set to `yes`, then `packet-redundancy=n/a`.

Location: voip{x x}>rt-fax-options

Configuring the fax data transmission rate

The Max-Rate parameter allows MultiVoice to modify the rate negotiation between the originating and destination fax terminals. This improves the reliability of the fax transmission by reducing the number of lost or repeated packets which occur during high rate transmissions, and reduces the required bandwidth for fax transmissions.

The fax transmission rate is regulated by modifying the content of the Digital Identification Signal (DIS) frame transmitted from the destination fax. Upon receipt of that DIS frame, the originating fax will use the data transmission rate specified by the Max-Rate parameter (or slower), and a corresponding modulation type. The content of the DIS frame is defined in the ITU Telecommunication sector standard (ITU-T) T.30, *Procedures for document facsimile transmission in general switched telephone networks*.

Changing the Max-Rate parameter modifies the high speed data transmission rate reported by the destination fax, and masks certain modulation types associated with higher fax transmission speeds. For example, when the data rate is set for 9600 bps, V.17 and V.33 are disallowed even though V.17 supports 9600 and 7200 bps. This implementation is used because:

- The DIS frame can specify only the supported modulation types for the highest selected transmission speeds at the destination fax,
- The calling fax terminal requires "training" to match the supported modulation.

Usage: Values assigned to the Max-Rate parameter cause MultiVoice to do the following:

Parameter value	Specifies
14400	Default. Mask the fax capabilities in the DIS frame that support fax data transmission at rates higher than 14,400 bps.
9600	Mask the fax capabilities in the DIS frame that support fax data transmission at rates higher than 9,600 bps.
4800	Mask the fax capabilities in the DIS frame that support fax data transmission at rates higher than 4,800 bps.
2400	Mask the fax capabilities in the DIS frame that support fax data transmission at rates higher than 2,400 bps.

The following example illustrates how to set the fax data transmission rates:

```
admin> read voip { 0 0 }
VOIP/{ 0 0 } read

admin> list rt-fax-options
[in VOIP/{ 0 0 }:rt-fax-options]

admin> set max-rate=9600

admin> write
VOIP/{ 0 0 } written
```

Dependencies: This parameter has the following dependencies:

- This parameter is N/A when `rt-fax-enable=no`.
- Changes made to this parameter are enabled for the next VoIP call.

Location: Voip{X X}->Rt-Fax-Options

Modified E1 profile settings in MAX TNT TAOS 8.0-103

The following parameters (shown with new values) are new or modified in MAX TNT TAOS 8.0-103:

```
[in E1/{ 1 1 5 }:line-interface]
signaling-mode = dtmf-r2-signaling
number-complete = 15-digits
```


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Parameter	Specifies
Signaling-Mode	This parameter has been enhanced to allow processing of Dual Tone Multi-Frequency (DTMF) tones over R2 signaling trunks by MultiVoice Gateways. This modification allows the MAX TNT to recognize and respond to either country specific R2 signaling (MFC-R2) or DTMF signaling over trunks supporting standard R2 signaling.
Number-Complete	This parameter has been enhanced to allow collection of up to 15 digits for R2 dial strings without waiting for end-of-pulse signaling.

Enabling DTMF-R2 signal processing

A new option added to the Signaling-Mode parameter allows MultiVoice Gateways to support DTMF R2 signaling generated by smaller European network switches and PBXs. MultiVoice implements DTMF tone processing using the R2 signaling standard defined by the International Telecommunications Union Telecommunication sector standard (ITU-T) Q.400, *Specifications of Signaling System R2 Definition and Function of Signals -- Forward Line Signals*.

To support DTMF-R2 detection, MultiVoice requires the following:

- Connection to E1 trunks attached to a switch that supports the ITU-T R2 signaling standard
- The switch must generate and/or relay the high-frequency/low-frequency tone combinations generated by normal touch tone dialing to the MultiVoice Gateway
- E1/R2 signaling must be enabled on the MultiVoice Gateway. This may be verified by checking the Base profile for the `r2-signaling-enabled=yes` entry

Detection of DTMF R2 signals is enabled from the E1 line profile.

DTMF tone detection

When processing tones for DTMF R2 signaling, the MultiVoice Gateway will:

- Upon detection of an inbound call, allocate a DSP for detecting DTMF tones; capturing DTMF digits as they are received from the switch.
- Upon receipt of an outbound call (from the packet network) allocate a DSP for generating DTMF tones; sending the first DTMF tone for 70ms, followed by 70ms of silence. This tone/silence sequence is repeated until all digits are sent to the telephone switch.

Usage: Setting the value of the Signaling-Mode parameter to `dtmf-r2-signaling` value enables the MAX TNT to recognize and respond to the DTMF R2 signal set during voice and data calls.

The following example illustrates how to enable multiple logical gateway processing on this MAX TNT:

```
admin> read e1 { 1 1 7 }
E1/{ 1 1 7 } read

admin> set signaling-mode=dtmf-r2-signaling

admin> write
E1/{ 1 1 7 } written
```

Dependencies: The following dependencies apply when `signaling-mode=dtmf-r2-signaling`:

- Once selected, DTMF R2 detection is enabled with the next VoIP call
- DTMF R2 detection is only supported when R2 signal processing is enabled for this MultiVoice Gateway.

Location: E1 { x x x }>Line-Interface

Collecting 15-digit dial strings

The Number-Complete parameter may now be used to configure a MAX TNT to collect 15-digit dial strings off of E1 trunks supporting inband CMF R2. This allows a MAX TNT to interoperate with European telephone systems that use E.164 addresses which are up to 15 digits long, without waiting for an end-of-pulse signal.

Previously, MultiVoice Gateways could be configured to collect dial strings of up to only 11 digits. For European networks using dial strings that were 12 digits or longer, a MultiVoice Gateway could only be configured to wait for the end-of-pulse signal to confirm it received all the dialed digits.

Usage: This parameter now accepts values from `0-digits` through `15-digits`, or `end-of-pulse` as valid entries.

The following example illustrates how to enable multiple logical gateway processing on this MAX TNT:

```
admin> read e1 { 1 1 7 }
E1/{ 1 1 7 } read
admin> set number-complete=15-digits
admin> write
E1/{ 1 1 7 } written
```

Dependencies: The following dependencies apply to this parameter:

- Changes are applied with the next VoIP call
- This parameter defaults to `N/A` when the Signaling-Mode parameter is assigned the following values:
  - `e1-kuwait-signaling`
  - `isdn`
  - `p7`
  - `dpnss`
  - `none`

Location: E1 { x x x }>Line-Interface

New VoIP profile settings in MAX TNT TAOS 8.0.1

The following parameters (shown with default values) are new or modified in MAX TNT TAOS 8.0.1:

```
[in VOIP/{ 0 0 }]
voice-ann-dir = /current
```

```
allow-g711-fallback = yes
allow-coder-fallback = yes
choose-dsp-via = voip-centric
trunk-quiesce-enable = no
early-ringback-enable = no
trunk-prefix-enable = no
```

Parameter	Specifies
Voice-Ann-Dir	Location of voice announcement files on a PCMCIA flash memory card in the MAX TNT unit. In previous releases, the value was read-only. In MAX TNT TAOS 8.0-103, administrators can create directories on the flash memory file system and specify a location for voice announcement files. See *Storing voice announcements in the FAT-16 flash memory file system* on page 36.
Allow-G711-Fallback	Enable/disable selection of the G.711 codec if the Gateway is unable to select its preferred codec. This parameter does not apply if Allow-Coder-Fallback is set to no. For details, see *Allowing fallback to alternate codecs* on page 37.
Allow-Coder-Fallback	Enable/disable selection of an alternate codec if the Gateway is unable to select its preferred codec. For details, see *Allowing fallback to alternate codecs* on page 37.
Choose-DSP-Via	*Not currently supported.*
Trunk-Quiesce-Enable	Enable/disable deactivation of a T1 PRI line when a Gateway is unavailable. For details, see *Deactivating trunks used for VoIP calls* on page 37.
Early-Ringback-Enable	Enable/disable generation of an early ringback tone on networks experiencing long call setup times. If the parameter is set to yes, the near-end Gateway plays a ringback tone to the caller as soon as a call connection is established with the far-end Gateway.
Trunk-Prefix-Enable	Enable/disable identification of the entry (ingress) trunk number to the exit (egress) Gateway or call signaling entity by prepending the ingress trunk number to the DNIS number.

Storing voice announcements in the FAT-16 flash memory file system

By default, MultiVoice callers are notified of call progress by DTMF-based tones. The tones report easily recognized call states such as ringback, busy signal, and so forth, as well as tones specific to MultiVoice, such as PIN prompt, which are not as easily recognized by callers. In previous MultiVoice releases, the MAX TNT introduced support for the playback of custom voice announcements to callers to indicate call progress. For details about how voice announcements work, and for information about managing them in the MAX TNT, see the *MultiVoice for the MAX TNT Configuration Guide* at http://www.ascend.com/doclibrary.

With MAX TNT TAOS 8.0-103, you can create directories on the flash memory file system and specify a location for voice announcement files. After creating the directory on a flash card and moving voice announcement files into it, specify the pathname in the Voice-Ann-Dir setting. For example, the following commands create a directory named messages and a subdirectory named announce on the flash card in slot 1:

```
admin> mkdir 1/messages

admin> mkdir 1/messages/announce
```

The following command loads a voice-announcement file named `busy.au` from a TFTP server at 10.10.10.10 to the `/current` directory on flash card 1 (flash card 1 is the default):

```
admin> load file network 10.10.10.10 busy.au
```

The following command moves the `busy.au` file to the new subdirectory on flash card 1:

```
admin> mv 1/current/busy.au 1/messages/announce/busy.au
```

The following commands inform the MultiVoice subsystem of the location of the voice announcement files:

```
admin> read voip { 0 0 }
VOIP/{ 0 0 } read

admin> set voice-ann-dir = /messages/announce

admin> write
VOIP/{ 0 0 } written
```

You can specify a pathname up to 40 characters long. When the system receives a request to play an announcement, it looks in the specified directory on the flash card in slot 1. If the card is not present or the voice announcement file is not found, the system looks for the specified directory on flash card 2.

Allowing fallback to alternate codecs

Voice is transmitted across an IP network as compressed audio frames. The Packet-Audio-Mode parameter in the default VoIP profile specifies the preferred audio codec used by the Gateways to compress and uncompress analog speech and digital audio frames.

In MAX TNT TAOS 8.0-103, you can set the following parameters (shown with default values) to specify how the system behaves when the preferred codec is not supported:

```
[in VOIP/{ 0 0 }]
allow-g711-fallback = yes
allow-coder-fallback = yes
```

Normally, an H.323 stack advertises a list of supported audio codecs. If the preferred codec is present on a list received from a far-end Gateway, that codec is always selected. Otherwise, the system selects an alternate codec that matches one from its supported list.

The Allow-Coder-Fallback parameter can be set to `no` to override the default system behavior and force the Gateway to reject the call if it is unable to select its preferred codec. If this parameter is set to `no`, the Allow-G711-Fallback parameter has no effect.

If Allow-Coder-Fallback parameter is set to `yes`, you can set the Allow-G711-Fallback parameter to `no` to prevent the system from selecting the G.711 codec when selecting an alternate codec. In this case, the system terminates the call if G.711 is the only available choice and it is not the preferred code. This setting affects VoIP, fax, and transparent modem calls.

Deactivating trunks used for VoIP calls

The trunk deactivation feature enables MultiVoice Gateways to automatically deactivate trunks used for VoIP calls when a Gateway becomes unavailable. This feature allows Gatekeepers in the MultiVoice network to route calls to other available Gateways, to use network resources more efficiently and improve service quality for users.

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Note: In this release, only T1 trunks that use ISDN PRI signaling and have been configured for VoIP can be deactivated system-wide by using this feature.

Trunk deactivation prevents the PSTN switch from routing subsequent calls to the trunks configured for VoIP. Current calls remain active until those calls are terminated by the caller or PSTN. When trunk deactivation is enabled, trunks configured to accept VoIP calls are made unavailable to the PSTN under the following conditions:

- A Gateway cannot register with either a primary or secondary Gatekeeper.
- A Gateway's trunk connection with the PSTN is unavailable, so that Gateway is forced to unregister itself from its Gatekeepers.

Previously, when a Gateway could not register with the primary and secondary Gatekeeper, the caller heard a fast busy signal because the PSTN switch continued to route calls to the trunks on that Gateway. Deactivating the trunk changes the trunk state to inform the PSTN switch aware that those trunks are not available.

Previously, when a VoIP call could not connect because a trunk was not operating, the caller heard a fast busy signal, because the Gatekeeper continued to route calls to that Gateway as long as it remained registered. Deactivating the trunk forces the Gateway to unregister from all known Gatekeepers, which causes the Gatekeepers to reroute new calls to other Gateways. When any one of the Gateway's trunks comes back in service, that Gateway starts registering itself with one of its known Gatekeepers. The Gatekeeper then begins to route calls to this Gateway.

The following commands enable trunk deactivation for T1 PRI lines configured for VoIP:

```
admin> read voip { 0 0 }
VOIP/{ 0 0 } read

admin> set trunk-quiesce-enable = yes

admin> write
VOIP/{ 0 0 } written
```

Enabling early ringback

For certain VoIP network configurations, such as satellite IP networks, wireless networks, or networks using channel-associated signaling (CAS) trunks, call setup times can be quite long. Callers might hang up before the call completes because they hear no call progress tones until RTP carries ringback from the far end PSTN. Early ringback allows the MAX TNT to generate a ringback tone locally, as soon as the call is started on the far-end Gateway.

Note: Early ringback is intended for use only on networks that experience long call setup times. Its use for other network configurations is not recommended, and might result in erroneous ring-to-busy and ring-to-failure announcements.

The following commands enable early ringback:

```
admin> read voip { 0 0 }
VOIP/{ 0 0 } read

admin> set early-ringback-enable = yes

admin> write
VOIP/{ 0 0 } written
```


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Trunk prefixing

Trunk prefixing enables the MAX TNT to identify the entry (ingress) trunk number to the exit (egress) gateway or call signaling entity by prepending the ingress trunk number to the DNIS number. Trunk groups must be in use system-wide.

When trunk prefixing is enabled, the system obtains the trunk group number of the ingress T1 trunk from the `trunk-group` setting in the T1 line profile, and prepends it to the detected DNIS number. The Q.931 Called Party Number information element (IE) in an H.225/Q.931 SETUP message then contains the DNIS number prefixed by the incoming trunk number. The destination address value of the SETUP user-to-user information element (UUIE) is not currently encoded.

For example, the following commands enable trunk prefixing, beginning with the next VoIP call the MAX TNT receives:

```
admin> read voip { 0 0 }
VOIP/{ 0 0 } read

admin> set trunk-prefix-enable = yes

admin> write
VOIP/{ 0 0 } written
```

Real-time fax

MultiVoice real-time fax uses the VoIP framework for call establishment, fax initiation, and detection of an incoming fax call.

Note: Real-time fax communications require guaranteed quality of service between the two fax-capable Gateways. The packet loss on the network must be less than 1%.

Real-time fax calls begin when a VoIP call is placed from an originating fax machine to the answering machine. If the MAX TNT is configured to perform out-of-band dual tone multifrequency (DTMF) signaling, a DSP automatically enables inband DTMF signaling at the start of the fax call. When the destination fax machine picks up the call and sends an answer tone, known as a CED tone, the destination Gateway detects this tone and initiates a switchover to real-time fax on both itself and the Gateway at the other end of the call. When the switchover is complete, the fax transmission proceeds normally.

You must create the appropriate coverage areas on the MultiVoice Access Manager to ensure that fax calls are routed between Gateways that are fax capable. For details, see the *MultiVoice Access Manager User's Guide* at http://www.ascend.com/doclibrary.

Overview of real-time fax settings

Following are the parameters (shown with default values) for enabling and improving the performance of real-time fax processing. Changes to these parameters take effect with the next VoIP call.

```
[in VOIP/{ 0 0 }:rt-fax-options]
rt-fax-enable = no
ecm-enable = yes
low-latency-mode = yes
command-spoof = yes
local-retransmit-lsf = yes
```


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Parameter	Specifies
RT-Fax-Enable	Enable/disable Real-time fax call processing. When the parameter is set to `no` (the default), fax tones are passed as if they were normal voice samples, and the other parameters in the subprofile are not applicable. When the parameter value is set to `yes`, this MAX TNT switches over from voice session to fax upon detection of a CED tone or V.21 HDLC flag.
ECM-Enable	Enable/disable error correction mode (ECM) for real-time fax calls. When the parameter is set to `yes` (the default), fax frames can be retransmitted in the event that a frame is not received correctly. ECM frames are relayed end to end between terminals. Setting the parameter to `no` disables ECM, so fax frames containing errors are not corrected.
Low-Latency-Mode	Enable/disable low latency mode for real-time fax operations over networks with low packet loss and low latency characteristics. Low latency mode allows operation on networks with less than 2.5 seconds or less of aggregate latency between pages. When the parameter is set to `no`, a minimum of 10 seconds delay is added to processing fax calls to allow interpretation of T.30 frames and implement spoofing.
Command-Spoof	Enable/disable spoofing of certain fax commands. Command spoofing is a method of improving performance and reducing fax errors on low latency networks.
Local-Retransmit-LSF	Enable/disable local retransmission of a low speed fax frame if no response is detected from the destination fax. This is designed to reduce fax transmission errors on low packet loss networks

In an SS7 environment, values in IPDC messages override corresponding call management settings in the default VoIP profile.

Example real-time fax configuration

For example, the following commands enable Real-time fax call processing and leave all performance parameters enabled:

```
admin> read voip { 0 0 }
VOIP/{ 0 0 } read

admin> set rt-fax-options rt-fax-enable = yes

admin> write
VOIP/{ 0 0 } written
```

Transparent modem

MultiVoice supports a transparent data mode that enables users to run a modem on a VoIP channel, regardless of the audio codec that is in use.

Overview of transparent modem settings

Following is the parameter for enabling the transparent modem features, shown with the default setting:

```
[in VOIP { 0 0 }]
g711-transparent-data = no
```

Parameter	Specifies
G711-Transparent-Data	Enable/disable transparent modem mode. When the parameter is set to yes, when the MAX TNT detects a modem in a VoIP channel, the unit transparently requests end-to-end G.711 encoding and bandwidth for the call, in a process similar to that used by real-time fax . The echo cancelers are disabled when the MAX TNT enters this mode, thus providing transparent G.711 encoding. The data is encoded transparently as an audio-mode type, either G.711 μ-law (64Kbps) or G.711 A-law (64Kbps). Settings take effect with the next incoming PSTN call. A separate license is not required for this feature.

In an SS7 environment, values in IPDC messages override corresponding call management settings in the default VoIP profile. For information about IPDC support for transparent modem, see *IPDC message support for fax and transparent modem* on page 43.

Example transparent modem configuration

The following commands enable the transparent modem feature on VoIP channels:

```
admin> read voip { 0 0 }
VOIP/{ 0 0 } read

admin> set g711-transparent-data = yes

admin> write
VOIP/{ 0 0 } written
```

Using transparent modem with real-time fax

If the MAX TNT has been licensed for real-time fax, users can run either a high-speed modem with speeds greater than 2400 bps or a fax terminal in the VoIP channel. This capability provides a fallback for real-time fax transmissions. Both fax terminals and high-speed modems transmit a single tone when they answer a call, but each type of equipment uses a different tone. The MAX TNT detects the type of equipment in use on the basis of its answer tone. When it detects the equipment answering the call, the MAX TNT sends H.245 request-mode messages to request a switchover from the current audio codec to either G.711 with no echo canceler (for transparent modem) or fax data mode (for real-time fax).

Transparent data is encoded as an audio-mode type, either G.711μ-law (64Kbps) or G.711 A-law (64Kbps). Real-time fax (if supported) is encoded as a fax data-mode type.

Note: Transparent data mode introduces an H.245 request-mode message that is not backward compatible with the real-time fax feature provided by previous MultiVoice releases. To interoperate with a Gateway using transparent mode, all Gateways must be upgraded to MAX TNT TAOS 8.0-103.

Example real-time fax and transparent modem configuration

The following commands enable both real-time fax and the transparent modem feature for high-speed modems:

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```
admin> read voip { 0 0 }
VOIP/{ 0 0 } read

admin> set rt-fax-options rt-fax-enable = yes

admin> set g711-transparent-data = yes

admin> write
VOIP/{ 0 0 } written
```

Limitation for low-speed modems

Real-time fax cannot be used concurrently with low-speed modems (2400bps or less) because these modems use the same answer tone as fax terminals. If a low-speed modem is used on a VoIP channel that is enabled for real-time fax, the Gateway detects a fax answer tone and requests T.38 encoding. The ingress Gateway (typically the Gateway on which the modem call originated) can accept the T.38 encoding request or reject the request, which causes the egress Gateway to terminate the call.

IPDC message support for modifying parameters

With MAX TNT TAOS 8.0-103, MAX TNT units provide limited support for IPDC messages used to modify the following values for VoIP calls. The request modify packet pass-through call (RMCP) message (`0x0015`) and accept modify packet pass-through call (AMCP) message (`0x0016`) allow modification of the following values for VoIP calls.

- VoIP encoding type (G.711µ-law, G.711A-law G.729, or G.723).
- Packet loading rate in frames per packet (value depends on VoIP encoding type)
- Source port type (currently, only the `SCN` value is supported).
- Destination port type (currently, only the `RTP` value is supported).
- Listen IP address.
- Listen RTP port number.
- Send IP address.
- Send RTP port number.

The MAX TNT can allocate its own system IP address as the listen IP address and RTP port and can specify its own send address and RTP port. For VoIP calls, you must avoid routing RTP traffic through the MAX TNT shelf controller. For that reason, when allowing the MAX TNT Gateway to allocate its own address, you must set the System-IP-Addr parameter in the IP-Global profile to an interface address other than the shelf-controller Ethernet port. For example, the following commands set the system address to the address of a port on an Ethernet card in slot 12:

```
admin> get ip-interface { { 1 12 1 } 0} ip-address
[in IP-INTERFACE/{ { shelf-1 slot-12 1 } 0 }:ip-address]
ip-address = 1.1.1.1/24

admin> read ip-global
IP-GLOBAL read

admin> set system-ip-addr = 1.1.1.1/24

admin> write
IP-GLOBAL written
```

In addition, you must make sure that VoIP calls can always find a route to the next-hop Gateway on the path to the destination VoIP Gateway. The route can be learned dynamically or configured as a static route. Many sites choose to configure default routes for VoIP traffic, so that RTP packets are never dropped due to lack of routing information. For example, the following commands configure a default route named VoIP to a next-hop Gateway at 2.2.2.2:

```
admin> new ip-route voip
IP-ROUTE/voip read

admin> set gateway = 2.2.2.2/24

admin> write
IP-ROUTE/VoIP written
```

IPDC message support for fax and transparent modem

Previously, transparent data for fax and modem calls was available only in an H.323 environment or for IPDC calls running G.711 codecs for VoIP. In this release, IPDC message request packet pass-through call (RCCP), accept packet pass-through call (ACCP), request modify for packet pass-through call (RMCP), and accept modify packet pass-through call (AMCP) messages enable an SS7 signaling gateway to direct the MAX TNT to enter T.38 fax mode or transparent modem mode on the basis of tone detection. In addition, the signaling gateway can control echo cancelation by disabling it or setting it to 32 milliseconds on a per-call basis.

The notify tone (NTN) message is used to notify the signaling gateway when an asynchronous fax or modem tone is detected. The MAX TNT sends this message to the signaling gateway if either fax or modem tone detection is enabled and the unit sees the tone. The MAX TNT detects fax tone if `rt-fax-enable` is set to `yes` in the default VoIP profile or if it receives the relevant IPDC message from the signaling gateway.

The MAX TNT detects modem tone if `g711-transparent-data` is set to `yes` in the default VoIP profile or if it receives the relevant IPDC message from the signaling gateway.

For an introduction to the real-time fax feature, see "Real-time fax," on page 39. For an introduction to the transparent modem feature, see "Transparent modem," on page 40.

New trunk features for VoIP calls

With MAX TNT TAOS 8.0.1, MAX TNT units provide a configurable timer for T1 lines that use inband signaling, a true connect feature to avoid charges for VoIP calls, and a calling line ID (CLID) generated by the MultiVoice Access Manager (MVAM).

Configurable interdigit timer for T1 inband signaling

When a T1 line uses inband signaling, you can enable Collect-Incoming-Digits to cause the DSP to decode the calling and called DTMF digits on the line, making DNIS and CLID information available for authentication and accounting. Following is the relevant parameter, shown with a sample setting:

```
[in T1/{ any-shelf any-slot 0 }:line-interface]
collect-incoming-digits = yes
```

In previous releases, when this feature was enabled, the T1 DSP always waited for 3 seconds after collecting the last digit before considering DNIS or automatic number identification

(ANI) collection complete. This 3-second timeout slowed down call setup times, and was unnecessary when a switch or PBX was generating the DTMF DNIS/ANI information with digit and interdigit times much smaller than 3 seconds. To improve call setup times, especially for VoIP calls with single-stage-dial, you can now configure the timeout for collecting incoming digits. Following is the relevant parameter, shown with its default value:

```
[in T1/{ any-shelf any-slot 0 }:line-interface]
t1-inter-digit-timeout = 3000
```

Parameter	Specifies
T1-Inter-Digit-Timeout	Number of milliseconds the T1 DSP waits between digits before considering DNIS/ANI collection complete. For backward compatibility, the default is 3 seconds. The valid range is 100 to 6000 milliseconds. The setting takes effect with the next incoming call. Specifying a lower value improves call setup times. This is especially important for VoIP calls with single-stage-dial. This parameter does not apply unless Collect-Incoming-Digits is set to `yes`.

For example, the following commands specify a timeout of half a second:

```
admin> read t1 { 1 2 3 }
T1/{ shelf-1 slot-2 3 } read

admin> set line-interface collect-incoming-digits = yes

admin> set line-interface t1-inter-digit-timeout = 500

admin> write
T1/{ shelf-1 slot-2 3 } written
```

Delaying charges until call is answered (true connect)

In earlier releases, incoming VoIP calls from the PSTN were connected at the near end Gateway before any H.323 signaling was sent to the far end Gateway. As a result, a PSTN charge was incurred at the time of connection to the near-end Gateway, before the called party received and answered the call from the far-end Gateway.

Now, you can change this behavior by enabling true connect. When this feature is enabled, alerting and connect messages sent to the PSTN switch are delayed to match the equivalent H.323 signaling to avoid incurring charges before a VoIP call has been answered.

The true connect feature requires a default call type of VoIP on T1 or E1 trunks accepting incoming VoIP calls. Following are the relevant parameters, shown with sample settings:

```
[in VOIP { 0 0 }]
true-connect-enable = yes

[in T1/{ shelf-1 slot-10 1 }:line-interface]
default-call-type = voip

[in E1/{ shelf-1 slot-11 1 }:line-interface]
default-call-type = voip
```

Parameter	Specifies
True-Connect-Enable	Enable/disable delay of PSTN alerting and connect messages to match the equivalent H.323 alerting and connect messages. The default setting is `no`, which results in the caller incurring a PSTN charge at the time of connection to the near-end Gateway, before the called party has received and answered the call from the far end Gateway. If set to `yes`, an alerting message is sent to the ingress PSTN switch only when an H.323 alerting message is received on the ingress VoIP Gateway. Similarly, a PSTN connect message is sent only when the H.323 VoIP call has been answered. This ensures that no charges are incurred for incomplete calls. The setting takes effect with the next incoming call. It has no effect on outbound calls.
Default-Call-Type	Must be set to VoIP for T1 or E1 trunks with incoming VoIP calls that require true connect. Note that setting this parameter to VoIP causes *all* calls received on the trunk to be mapped to VoIP.

For example, the following commands enable delayed PSTN alerting and connect messages on trunk lines configured with a default VoIP call type:

```
admin> read voip { 0 0 }
VoIP { 0 0 } read

admin> set true-connect-enable = yes

admin> write
VoIP { 0 0 } written
```

Note: For ISDN trunks, Lucent recommends that you set the T310 timer on the telephone company switch or PBX to 30 seconds or greater when using the true connect feature. because the T310 timeout value includes the time that the called party's telephone is ringing, a 10-second timeout can cause the near-end Gateway to disconnect the call too soon.

When the true connect feature is enabled and a VoIP call fails before the PSTN call is fully connected, the Gateway is still able to send an appropriate tone or voice announcement to the caller.

Gatekeeper CLID substitution

When MultiVoice Gateways are connecting VoIP calls, they can transmit a calling line ID (CLID) generated by the MVAM software on the Gatekeeper instead of the PSTN-generated CLID collected on the trunk line. CLID substitution allows the MultiVoice network to provide the appropriate E.164 address for both the called and calling telephone numbers to the respective PSTN, and for use by external applications.

In certain configurations in which the Gateways connecting the call reside in different area codes or countries, the CLID received from the PSTN must be changed to provide the appropriate calling number information to the local carrier, or to call management and billing applications.

When the MVAM receives the CLID from a Gateway, it translates the CLID to the appropriate dial string, adding or removing country codes and area codes as appropriate for the respective locations of the callers. The Gatekeeper then reports the revised CLID to the Gateways as part of the admission confirmed (ACF) message.

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